WHAT IS CLAIMED IS:

1. A method for the variable rate coding of a speech signal, comprising the steps of:

(a) classifying the speech signal as either active or inactive;

(b) classifying said active speech into one of a plurality of types of active speech;

(c) selecting a coding mode based on whether the speech signal is active or inactive, and if active, based further on said type of active speech; and

(d) encoding the speech signal according to said coding mode, forming an encoded speech signal.

2. The method of claim 1, further comprising the step of decoding said encoded speech signal according to said coding mode, forming a synthesized speech signal.

3. The method of claim 1, wherein said coding mode comprises a CELP coding mode, a PPP coding mode, or a NELP coding mode.

4. The method of claim 3, wherein said step of encoding encodes according to said coding mode at a predetermined bit rate associated with said coding mode.

5. The method of claim 4, wherein said CELP coding mode is associated with a bit rate of 8500 bits per second, said PPP coding mode is associated with a bit rate of 3900 bits per second, and said NELP coding mode is associated with a bit rate of 1550 bits per second.

6. The method of claim 3, wherein said coding mode further comprises a zero rate mode.

7. The method of claim 1, wherein said plurality of types of active speech include voiced, unvoiced, and transient active speech.

8. The method of claim 7, wherein said step of selecting a coding mode comprises the steps of:

(a) selecting a CELP mode if said speech is classified as active transient speech;

(b) selecting a PPP mode if said speech is classified as active voiced speech; and

(c) selecting a NELP mode if said speech is classified as inactive speech or active unvoiced speech.

9. The method of claim 8, wherein said encoded speech signal comprises codebook parameters and pitch filter parameters if said CELP mode is selected, codebook parameters and rotational parameters if said PPP mode is selected, or codebook parameters if said NELP mode is selected.

10. The method of claim 1, further comprising the step of calculating initial parameters using a "look ahead."

11. The method of claim 10, wherein said initial parameters comprise LPC coefficients.

12. The method of claim 1, wherein said coding mode comprises a NELP coding mode, wherein the speech signal is represented by a residual signal generated by filtering the speech signal with a Linear Predictive Coding (LPC) analysis filter, and wherein said step of encoding comprises the steps of:

(i) estimating the energy of the residual signal, and

(ii) selecting a codevector from a first codebook, wherein said codevector approximates said estimated energy;

and wherein said step of decoding comprises the steps of:

(i) generating a random vector,

(ii) retrieving said codevector from a second codebook,

(iii) scaling said random vector based on said codevector, such that the energy of said scaled random vector approximates said estimated energy, and

(iv) filtering said scaled random vector with a LPC synthesis filter, wherein said filtered scaled random vector forms said synthesized speech signal.

13. The method of claim 12, wherein the speech signal is divided into frames, wherein each of said frames comprises two or more subframes, wherein said step of estimating the energy comprises the step of estimating the energy of the residual signal for each of said subframes, and wherein said codevector comprises a value approximating said estimated energy for each of said subframes.

14. The method of claim 12, wherein said first codebook and said second codebook are stochastic codebooks.

15. The method of claim 12, wherein said first codebook and said second codebook are trained codebooks.

16. The method of claim 12, wherein said random vector comprises a unit variance random vector.

17. A variable rate coding system for coding a speech signal, comprising:

classification means for classifying the speech signal as active or inactive, and if active, for classifying the active speech as one of a plurality of types of active speech; and

a plurality of encoding means for encoding the speech signal as an encoded speech signal, wherein said encoding means are dynamically selected to encode the speech signal based on whether the speech signal is active or inactive, and if active, based further on said type of active speech.

18. The system of claim 17, further comprising a plurality of decoding means for decoding said encoded speech signal.

19. The system of claim 17, wherein said plurality of encoding means includes a CELP encoding means, a PPP encoding means, and a NELP encoding means.

20. The system of claim 18, wherein said plurality of decoding means includes a CELP decoding means, a PPP decoding means, and a NELP decoding means.

21. The system of claim 19, wherein each of said encoding means encodes at a predetermined bit rate.

22. The system of claim 21, wherein said CELP encoding means encodes at a rate of 8500 bits per second, said PPP encoding means encodes at a rate of 3900 bits per second, and said NELP encoding means encodes at a rate of 1550 bits per second.

23. The system of claim 19, wherein said plurality of encoding means further includes a zero rate encoding means, and wherein said plurality of decoding means further includes a zero rate decoding means.

24. The system of claim 17, wherein said plurality of types of active speech include voiced, unvoiced, and transient active speech.

25. The system of claim 24, wherein said CELP encoder is selected if said speech is classified as active transient speech, wherein said PPP encoder is selected if said speech is classified as active voiced speech, and wherein said NELP encoder is selected if said speech is classified as inactive speech or active unvoiced speech.

26. The system of claim 17, wherein said encoded speech signal comprises codebook parameters and pitch filter parameters if said CELP encoder is selected, codebook parameters and rotational parameters if said PPP encoder is selected, or codebook parameters if said NELP encoder is selected.

27. The system of claim 17, wherein the speech signal is represented by a residual signal generated by filtering the speech signal with a Linear Predictive Coding (LPC) analysis filter, and wherein said plurality of encoding means includes a NELP encoding means comprising:

energy estimator means for calculating an estimate of the energy of the residual signal, and

encoding codebook means for selecting a codevector from a first codebook, wherein said codevector approximates said estimated energy;

and wherein said plurality of decoding means includes a NELP decoding means comprising:

random number generator means for generating a random vector,

decoding codebook means for retrieving said codevector from a second codebook,

multiply means for scaling said random vector based on said codevector, such that the energy of said scaled random vector approximates said estimate, and

means for filtering said scaled random vector with an LPC synthesis filter, wherein said filtered scaled random vector forms said synthesized speech signal.

28. The system of claim 17, wherein the speech signal is divided into frames, wherein each of said frames comprises two or more subframes, wherein said energy estimator means calculates an estimate of the energy of the residual signal for each of said subframes, and wherein said codevector comprises a value approximating said subframe estimate for each of said subframes.

29. The system of claim 17, wherein said first codebook and said second codebook are stochastic codebooks.

30. The system of claim 17, wherein said first codebook and said second codebook are trained codebooks.

31. The system of claim 17, wherein said random vector comprises a unit variance random vector.